CLAIMS:

1. A method of generating a monaural signal (S) comprising a combination of at least two input audio channels (L, R), comprising the steps of:

for each of a plurality of sequential segments (t(n)) of said audio channels (L,R), summing (46) corresponding frequency components from respective frequency spectrum representations for each audio channel (L(k), R(k)) to provide a set of summed frequency components (S(k)) for each sequential segment;

for each of said plurality of sequential segments, calculating (45) a correction factor (m(i)) for each of a plurality of frequency bands (i) as function of the energy of the frequency components of the summed signal in said band ($\sum_{k \in i} |S(k)|^2$) and the energy of said frequency components of the input audio channels in said band ($\sum_{k \in i} \{|L(k)|^2 + |R(k)|^2\}$); and

correcting (47) each summed frequency component as a function of the correction factor (m(i)) for the frequency band of said component.

2. A method according to claim 1 further comprising the steps of:

providing (42) a respective set of sampled signal values for each of a plurality of sequential segments for each input audio channel; and

for each of said plurality of sequential segments, transforming (44) each of said set of sampled signal values into the frequency domain to provide said complex frequency spectrum representations of each input audio channel (L(k),R(k)).

3. A method according to claim 2 wherein the step of providing said sets of sampled signal values comprises:

for each input audio channel, combining overlapping segments (m1,m2) into respective time-domain signals representing each channel for a time window (t(n)).

4. A method according to claim 1 further comprising the step of:

for each sequential segment, converting (48) said corrected frequency spectrum representation of said summed signal (S'(k)) into the time domain.

5. A method according to claim 4 further comprising the step of: applying overlap-add (50) to successive converted summed signal representations to provide a final summed signal (s1,s2).

6. A method according to claim 1 wherein two input audio channels are summed and wherein said correction factors (m(i)) are determined according to the function:

$$m^2(i) = \frac{\sum_{k\in i}\left\{|L(k)|^2 + |R(k)|^2\right\}}{2\sum_{k\in i}|S(k)|^2} = \frac{\sum_{k\in i}\left\{|L(k)|^2 + |R(k)|^2\right\}}{2\sum_{k\in i}|L(k)+R(k)|^2}$$

7. A method according to claim 1 wherein two or more input audio channels (X_n) are summed according to the function:

$$S(k) = C(k)\sum_n w_n(k)X_n(k)$$

wherein C(k) is the correction factor for each frequency component and wherein said correction factors (m(i)) for each frequency band are determined according to the function:

$$m^2(i) = \frac{\sum_n \sum_{k\in i}|w_n(k)X_n(k)|^2}{n\sum_{k\in i}\left|\sum_n w_n(k)X_n(k)\right|^2}$$

wherein $w_n(k)$ comprises a frequency-dependent weighting factor for each input channel.

8. A method according to claim 7 wherein $w_n(k)=1$ for all input audio channels.

9. A method according to claim 7 wherein $w_n(k)\neq 1$ for at least some input audio channels.

10. A method according to claim 7 wherein the correction factor for each frequency component (C(k)) is derived from a linear interpolation of the correction factors (m(i)) for at least one band.

11. A method according to claim 1 further comprising the steps of:

for each of said plurality of frequency bands, determining an indicator ($\alpha(i)$) of the phase difference between frequency components of said audio channels in a sequential segment; and

prior to summing corresponding frequency components, transforming the frequency components of at least one of said audio channels as a function of said indicator for the frequency band of said frequency components.

12. A method according to claim 11 wherein said transforming step comprises operating the following functions on frequency components (L(k), R(k)) of left and right input audio channels (L,R):

$$L'(k) = e^{jc\alpha(i)} L(k)$$
$$R'(k) = e^{-j(1-c)\alpha(i)} R(k)$$

wherein $0 \leq c \leq 1$ determines the distribution of phase alignment between the said input channels.

13. A method according to claim 1 wherein said correction factor is a function of a sum of energy of the frequency components of the summed signal in said band and a sum of the energy of said frequency components of the input audio channels in said band.

14. A component (S8') for generating a monaural signal from a combination of at least two input audio channels (L, R), comprising:

a summer (46) arranged to sum, for each of a plurality of sequential segments (t(n)) of said audio channels (L,R), corresponding frequency components from respective frequency spectrum representations for each audio channel (L(k), R(k)) to provide a set of summed frequency components (S(k)) for each sequential segment;

means for calculating (45) a correction factor (m(i)) for each of a plurality of frequency bands (i) of each of said plurality of sequential segments as function of the energy of the frequency components of the summed signal in said band ($\sum_{k \in i} |S(k)|^2$) and the energy of said frequency components of the input audio channels in said band ($\sum_{k \in i} \{|L(k)|^2 + |R(k)|^2\}$); and

a correction filter (47) for correcting each summed frequency component as a function of the correction factor (m(i)) for the frequency band of said component.

15. An audio coder including the component of claim 14.

16. Audio system comprising an audio coder as claimed in claim 15 and a compatible audio player.